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LEXINGTON MASSACHUSETTS

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ABSTRACT

The flashing light encoder has the assignment of transforming data information imposed on a group of FM sub-carrier channels into a form compatable with a flashing light transmitter. The encoder samples the frequency in X-channels and converts each frequency into a time interval between two flashes. The coincidence of the start of one sample with the termination of the previous sample provides for maximum efficienty in the utilization of time and power.

TABLE OF CONTENTS

FIGURES

FLASHING LIGHT ENCODERS

INTRODUCTION

Need For Flashing Light Encoders

Transmission of data from sub-orbital vehicles is accomplished primarily through the use of RF telemetry methods. However, during certain phases of a vehicle's re-entry the build up of plasma sheaths around a vehicle is such as to prohibit RF transmission from the vehicle proper. One approach that has been suggested to assure data transmission during the RF blackout employs a supplementary flashing light transmitting system. This technique is illustrated in Fig. 1. By either having the flashing light transmitter operate continually or operate only during the RF blackout a continuous flow of data can be realized.

For best economy in power the light transmission is in the form of very short, high peak power light pulses with the intervals between pulses being determined by the data being transmitted. As shown in Fig. 1 it is the encoder which must transform the data being transmitted into a form that can be used by the flashing light system.

Some general approaches to the design of flashing light encoders are discussed in this report; and the one channel and multichannel encoders are investigated in detail. This investigation was an out-growth of the Trailblazer II program conducted by Group 312 of Lincoln Laboratory.

GENERAL CONSIDERATIONS

Inputs To The Encoder

Since the flashing light transmission system is but a support system, the data to be transmitted is in a form usable by the RF telemetry system. Consequently, the data inputs to the flashing light

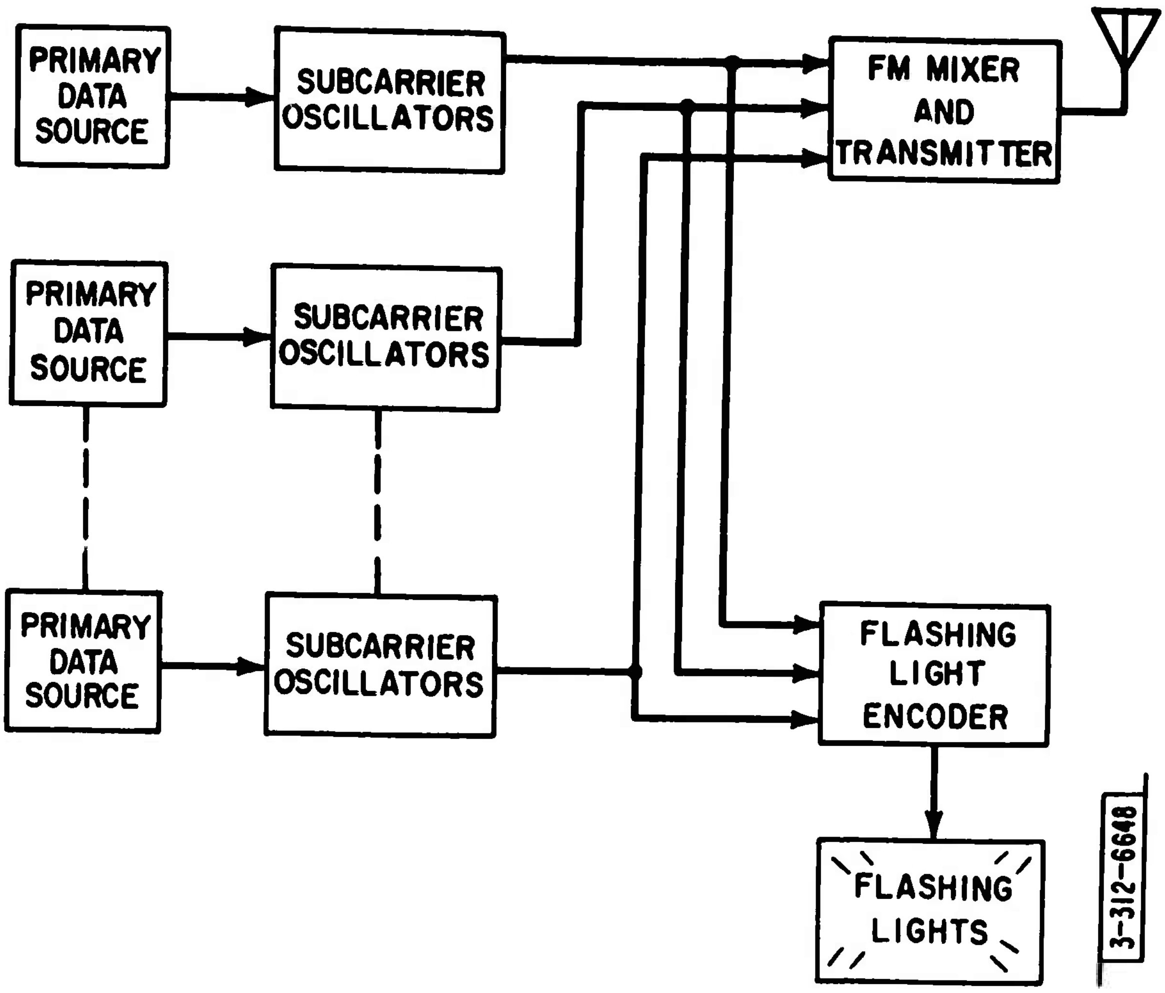

Figure 1. Role of Flashing Light Encoder.

encoder are in the form of standard FM telemetry subcarrier bands. Each data channel consists of a standard center frequency with a $\pm$ 7.5 % or $\pm$ 15 % deviation. The number of channels handled by the flashing light system may not be the same as that of the RF system: for all the data normally transmitted may not be important during the RF blackout. The generalized input to the encoder therefore consists of X channels of the standard subcarrier bands where X is equal to or less than the total number of subcarrier bands used in the RF transmission.

Outputs of the Encoder

The best form of output to save power and to retain high signal levels consists of a series of very short, high intensity, low duty cycle pulses of light. The output information is encoded into the pulse frequency or more precisely into the time interval between each two pulses. An economical form of encoding is illustrated in Fig. 2 Note that at the beginning of each cycling of data a signature pulse is necessary to establish a relative time reference. In this figure this is accomplished by transmitting a doublet whose width or time interval is less than the width of any data channel. The first interval following the doublet is denoted as T_A and contains data as encoded from input channel A. The next interval T_B contains the data from input channel B and the rest of the intervals follow suit. At the end of the final interval T_X the doublet is produced and the cycling of the data channels is recommenced. The interval of the Nth channel T_N varies between a minimum $T_N = T_1$ which is greater than the doublet interval and a maximum $T_N = T_2$ determined by the over-all cycling rate desired. If all X of the maximum intervals were approximately the same and if the entire cycling of data was to be completed in no greater than Y seconds then:

$$T_2 \leq \frac{Y}{X} \tag{1}$$

Note, that because each interval of time commences with the termination of the previous interval, the average interval will be less than T_2

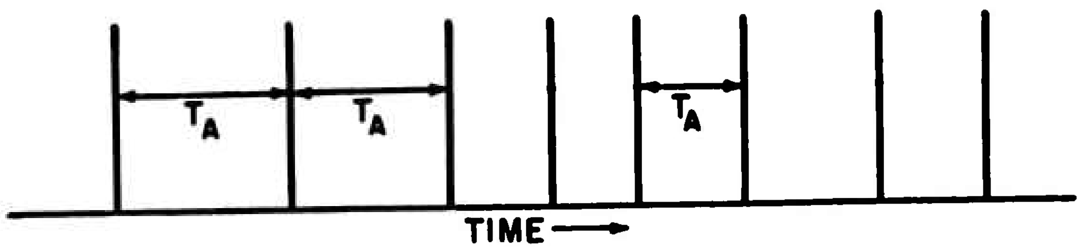

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Figure 2a. Train of Output Flashes from Flashing Lights for Single Channel System.

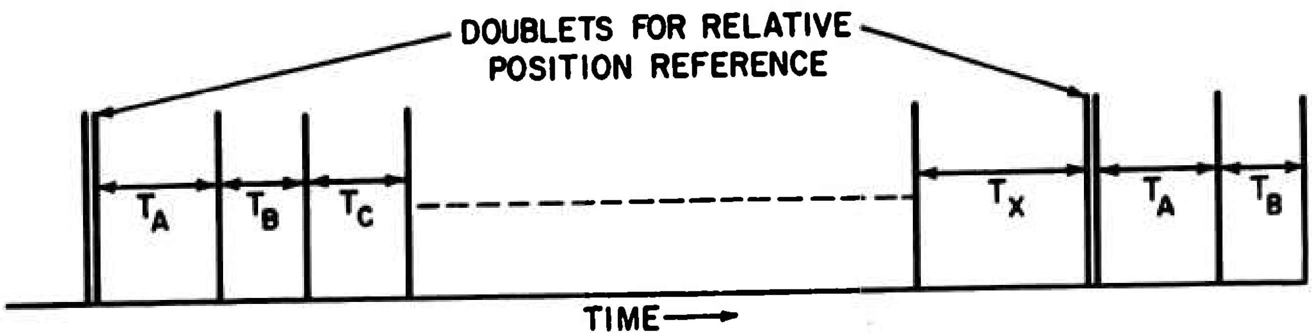


Figure 2b. Train of Output Flashes from Flashing Lights for an X Channel System.

and, in fact, if the encoder is a linear operator and the original data is linear with the subcarrier frequency, then as shown in Fig. 3A:

$$f_{N_{ave}} = \frac{f_{N_{max}} + f_{N_{min}}}{2} \tag{2}$$

or

$$T_{N_{ave}} = \frac{1}{f_{N_{ave}}} = 2\frac{T_1 T_2}{T_1 + T_2}, \quad 2T_1 > T_{ave} > T_1 \tag{3}$$

and since

$$T_{N_{ave}} - \frac{T_2}{2} < 0 \tag{4}$$

for $T_2 \geq 3T_1$

$$T_{ave} \leq \frac{Y}{2X} \tag{5}$$

therefore, if T_2 is 3 or more times larger than T_1 the average cycling time of data will take less than half the upper limit of Y seconds.

The ratio of $(T_2/T_1 = R)$ appears several times in the analysis of the generalized encoder. The ratio of dead space to used space is:

$$\frac{\text{DEAD SPACE}}{\text{USED SPACE}} = \frac{T_1}{T_2 - T_1} = \frac{1}{R - 1} \tag{6}$$

It would appear from this criterion that to use the time most efficiently it would be desirable to have R as large as possible. However, if the encoder is a linear device in frequency as shown in Fig. 3A, then the information is all crowded near T_1 for large R. This crowding reduces resolution of data. Therefore, some intermediate value of R should be sought. A range of R = 3 was selected as a reasonable design criterion. This keeps the ratio of $\frac{df}{dT}_{f = f_2} \Big/ \frac{df}{dT}_{f = f_1}$ to a range of three which helps simplify the ultimate reduction of data.

It should be mentioned before leaving this topic that this form of output is analogous to pulse width coding with the start and end of each pulse marked with a light flash. The approach discussed here, however, is more efficient in both conservation of usable space and conservation power. This is brought about by the fact that the flash marking the end of one pulse width also marks the beginning of the next pulse width. Only half as many pulses are needed and the dead space is greatly reduced.

Encoding

The flashing light encoder must perform two basic operations to transform the general input into the output previously discussed, namely subtraction of frequency and division of frequency. This first operation is necessary to make the deviation from center frequency the larger part of the signal. The second operation of division is needed to reduce this deviation in frequency to a value consistant with the flashing lights. These operations may be performed singularly or together by analog, digital or hybr means. Since the generalizing of nonlinear encoding sheds little light on the technique of synthesizing an encoder, this section will deal with linear encoding only. A particular nonlinear analog encoder, however, is discussed under multichannel encoders. Its form of output interval as function of frequency is illustrated in Fig. 3B.

The typical input to the encoder will be of the form of:

$$(f_o - \Delta f_o) \leq f \leq (f_o + \Delta f_o) \tag{7}$$

where f_o is the center frequency, Δf_o is the maximum deviation from center frequency (usually 0.075 f_o), and f is the instantaneous input frequency. With the subtraction of a constant frequency f_c and with frequency division by K the output becomes the reciprocal of the time interval as shown by Equation 8.

$$\frac{f - f_c}{K} = \frac{1}{T} \tag{8}$$

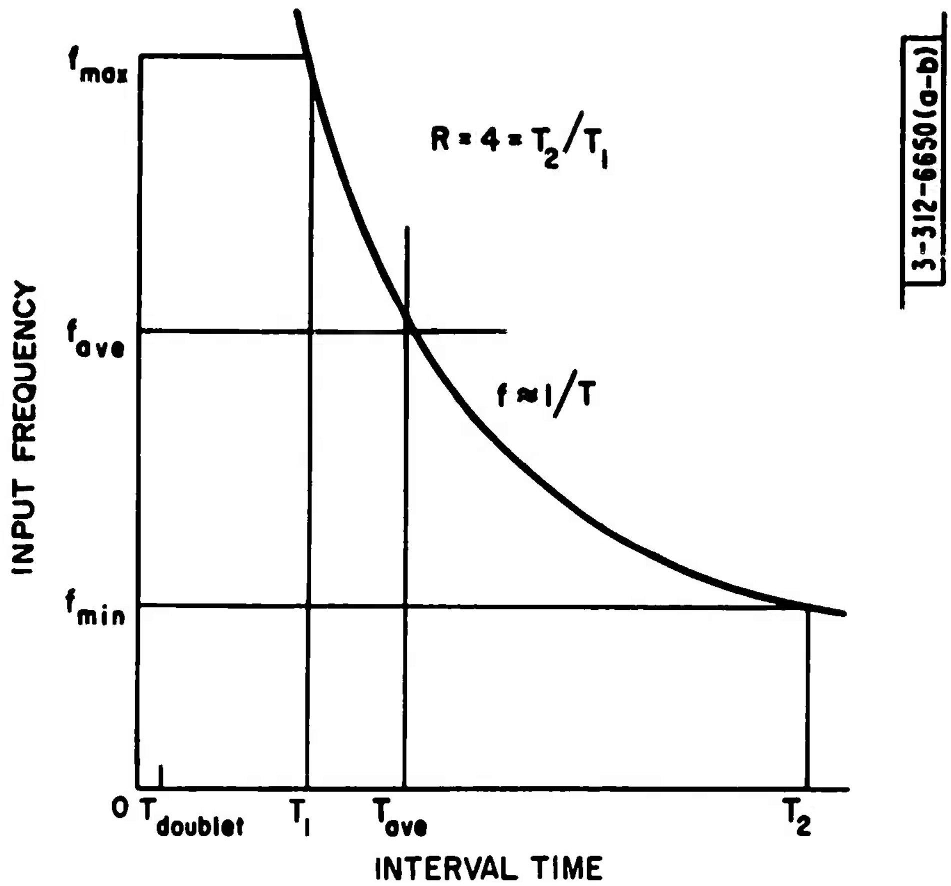


Figure 3a. Input Frequency vs. Interval Time for Linear Encoder (Digital).

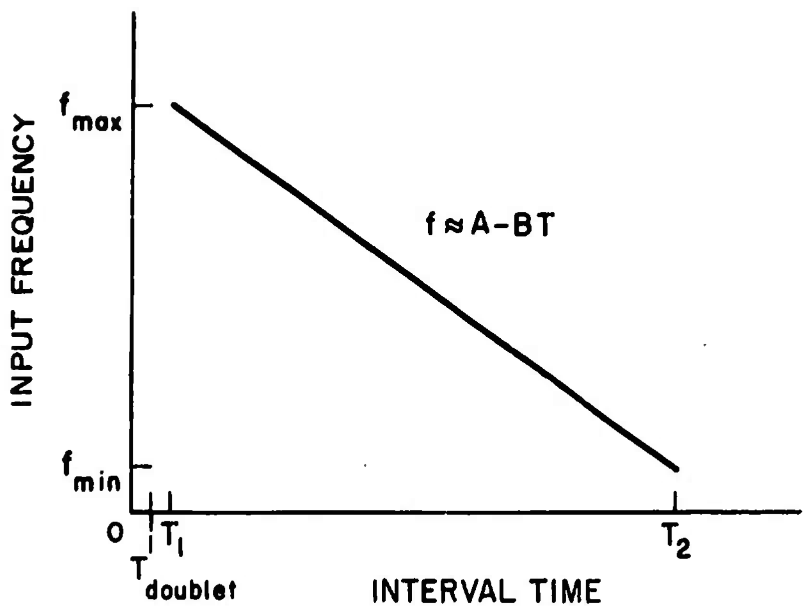


Figure 3b. Input Frequency vs. Interval Time for Special Nonlinear Encoder (Analog).

$$f_c > f_o + \Delta f_o \qquad \text{or} \qquad f_c < f_o - \Delta f_o \tag{9}$$

Note that f_c can either be higher or lower than but not in the range of f or a double valued output function will result. Once the f_o is known the two constants, K and f_c can be determined by the relation:

$$\frac{f_o + \Delta f_o - f_c}{K} = \pm \frac{1}{T_1} \tag{10A}$$

$$\frac{f_o - \Delta f_o + f_c}{K} = \pm \frac{1}{T_2} \tag{10B}$$

where T_1 and T_2 are determined from considerations discussed in the previous section. For digital and hybrid approaches it is convenient to have K a digit of high factorability and for maximum simplicity to have K equal to 2^L where L is a positive digit to be determined. With this requirement a solution for Eq. (10) would not in general be possible without some relaxation in the value of T_2 and T_1. From the previous section it was found that T_2/T_1 or R should be near 3 in value or $R = 3 + \Delta R$. Eliminating f_c from equation 10A and B and making the necessary substitutions we find:

$$\frac{3\ T_2}{(20)\ (2^L)} + 1 = R = 3 + \Delta R \tag{11A, B}$$

$$f_c = (.925) f_o\ (2^L) \pm \frac{2^L}{T_2}$$

Figure 4 is a graph of equation (11)A with R assumed to be three, Once a nondigit value of L has been determined from this graph the nearest or most convenient digit value for L is chosen and ΔR calculated from equation (12).

$$\Delta R = -1.386\ \Delta L \tag{12}$$

where ΔL is the difference between the digital and non-digital values of L.

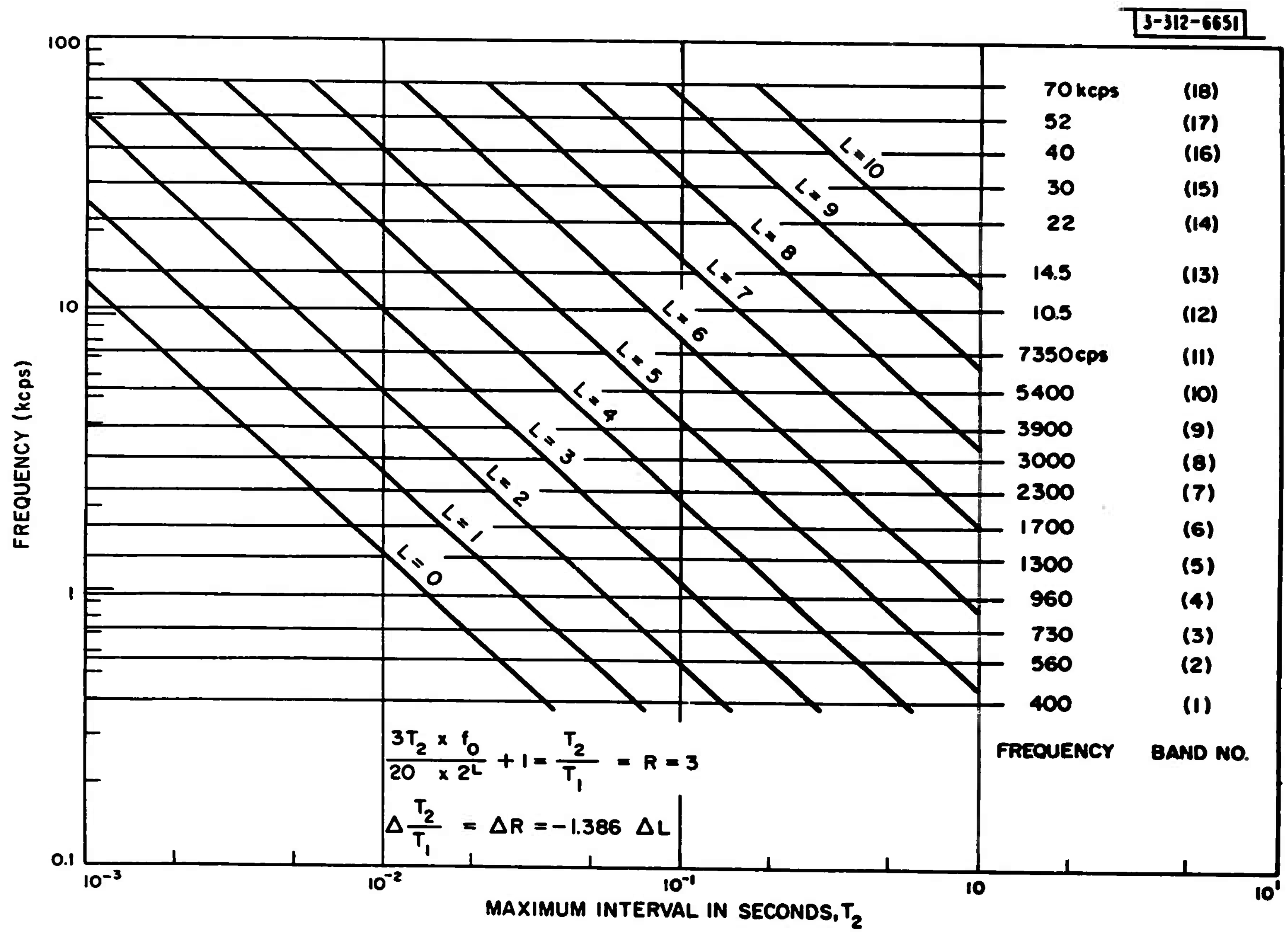


Figure 4. Determination of Number of Dividing Flip-Flops.

To Use this graph the following approach is recommended:

Steps in Determining L	Example
1. determine T_2 by Eq. (5)	3 channels, 100 ms cycle time $T_2 = 33$ ms
2. Determine subcarrier frequency to be used	13 14.5 Kc
3. find point of intersection as determined by graph I The point will occur between two values of L	$5 \leq L \leq 6$ $L = 5.2$
4. Use the equation $\Delta R = -1.386\Delta L$ to determine which of the two values of L should be selected. This will be the number of flip-flops needed for frequency division. For multichannel the larger value of L is preferred.	$2L = -0.2$ $R + \Delta R = 3.3$ $L = 5$
5. Solve equation (11)b for f_c, the frequency to be subtracted.	$f_c = 12.4$ Kc

Once f_c and 2^L have been determined for all channels the preliminary design has been completed. It may be desired that the same f_c be shared by two adjacent channels. Under these circumstances a further relaxation in R may be necessary.

Approaches

Of the many approaches investigated two seemed better suited for this application, the digital approach of Fig. 5 and the analog approach of Fig. 6. Both utilize the same technique for performing the initial subtraction in frequency. A local oscillator is mixed with the input signal through the use of a gate which, after low pass filtering, results in the difference frequency $(f - f_c)$. The subtraction in frequency is particularly critical, for if an over-all accuracy of $\epsilon\%$ is desired, the accuracy of the local oscillator must be better than $(.15)\epsilon\%$

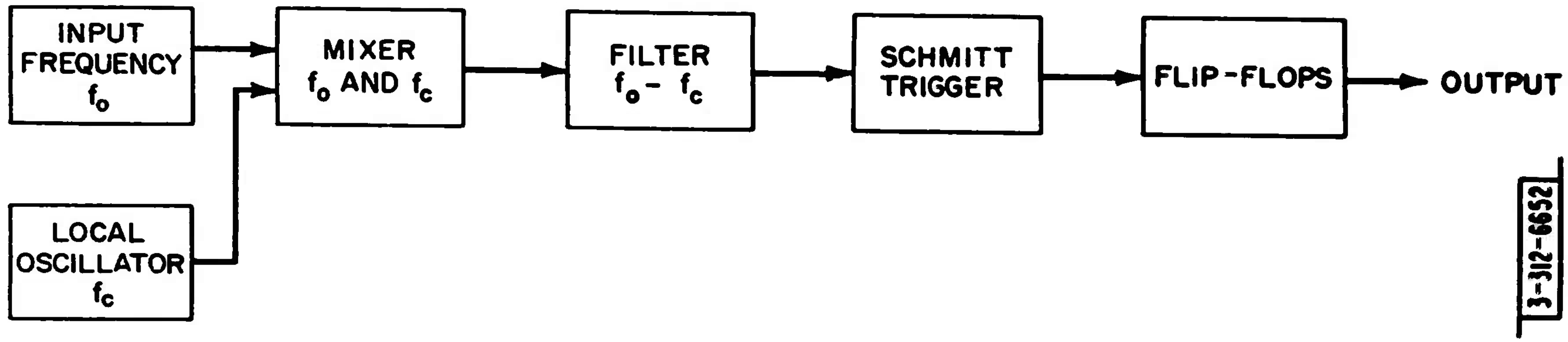


Figure 5. Digital Approach to Encoder.

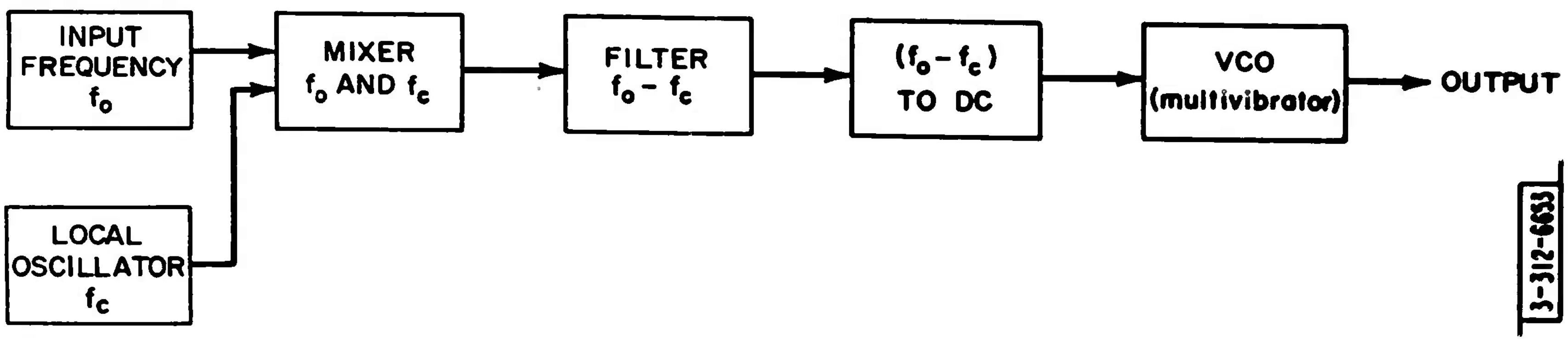


Figure 6. Analog Approach to Encoder.

since only 15 % of the input frequency spread is used for data. For this reason the local oscillator must be very stable.

The digital and analog approaches differ in their method of frequency division. In the digital approach the difference frequency sine wave is transformed into a square wave of fast rise time through the use of a Schmitt trigger circuit. This square wave is then frequency divided by 2^L where L is the number of flip-flops used in the frequency division.

The analog system converts the difference frequency to a dc voltage, the level of which is a function of the difference frequency. This dc voltage in turn controls a voltage controlled multivibrator. The analog system is subjected to error in frequency division due to the dc voltage conversion anu in the conversion of the dc voltage to the period of the multivibrator. The total error in the analog approach can be approximated by equation (13).

$$\text{Analog error} = 7\,(\text{osc error}) + (\text{freq to dc error}) + (\text{dc to } \tau \text{ error}) = \epsilon_A \qquad (13)$$

The one channel digital encoder has essentially no error introduced in addition to the local oscillator error and is therefore more accurate and hence more suitable than the analog system. However, for a multichannel encoder an inherent error is introduced that is independent of circuit components. If frequency division is initiated at a random time determined by the output of the $(N - 1)^{th}$ channel, and if L flip-flops are used for frequency division, then an output will be achieved after 2^L to $(2^L + 1)$ cycles. The output time then could be as much as $\frac{1}{2^L} \times 100\,\%$ too large which indicated a maximum error from mean of $\frac{1}{2^{(L+1)}} \times 100\%$. Since the sampling is not begun at random in the one channel encoder it does not suffer from this form of error.

For fixed cycle time of Y seconds L must necessarily become smaller as the number of channels become larger. Thus the larger the number of channels the greater the inherent error.

$$\text{max, digital error} = \left[7(\text{osc error}) + \frac{1}{2^{(L+1)}}\right] 100\ \% = \epsilon_{d_x} \quad (14)$$

for X channels $X > 1$

$$\text{max digital error} = 7(\text{osc error})\ 100\% = \epsilon_{d_1} \quad (15)$$

for one channel $X = 1$

therefore when:

$$\frac{1}{2^{L+1}} > (\text{freq. to dc error}) + (\text{dc to } \tau \text{ error}) \quad (16)$$

then the analog approach becomes more suitable from the stand point of error. Graph I, Fig. (4) will yield an approximate value for L and thus an approximate value for the inherent error in multichannel digital division.

The analog approach has another advantage which may override a decision based upon the above equation when accuracy is not critical. The output interval, T_N varies as $1/f$ in the digital system where as in the analog system T_N can be made to vary directly with f. This implies that the time intervals between flashes in the analog system could be made linear with the original data where as the intervals are linear with the inverse of the data or input frequency in the digital system. When the output interval is a linear function of input frequence there is no restriction on $R = \frac{T_2}{T_1}$ and therefore the dead space is eliminated.

The choice of approach then rests on the relative values attributed to accuracy, linearity, data rate, and number of channels desired.

A One Channel Working Model

The circuit of Fig. 7 is a one channel (channel 13) digital encoder. The encoder reduces the 14.5 kc $\pm$ 1.1 kc to intervals between flashes of from 32 to 10 milliseconds. Five flip-flops were used for frequency division as derived from the Graph of Fig. 4

The local oscillator was set at 12.4 kc as determined by equation 11 B. Temperature stablization was obtained through the use of diodes in each collector of the oscillator. A maximum drift from mean of 25 cycles or less than $\pm$ .2 % was achieved over the temperature range of from -20^{o} to $+ 160^{o}$C.

Mixing was performed by driving a gate alternately from cutoff to saturation thus obtaining, upon filtering, a difference frequency of maximum signal strength. The Schmitt trigger circuit is used to convert the difference frequency to a square wave and the five flip-flops divide the difference frequency by 32.

The sixth flip-flop and duo-output amplifier are employed to meet the special requirements of two outputs at half frequency and π radians out of phase with each other.

The entire circuitry was packaged in a 4"x 2"x 1"box and performed with a small error in a temperature range of from -68^{o}C to $+170^{o}$C (-90^{o}F to $+ 340^{o}$F).

The Multichannel Flashing Light Encoder

Shown in Fig. (8) is an eight channel encoder operating on input subcarrier bands 9-16. Up to and including the filters this design is primarily the same as that of the single channel digital encoder. If the ratio of the center frequencies of adjacent channels were equal to the square root of two then all local oscillators, each serving two channels, could be related by factors of two in frequency to a master oscillator. In general, however, the ratios of center frequencies of neighboring

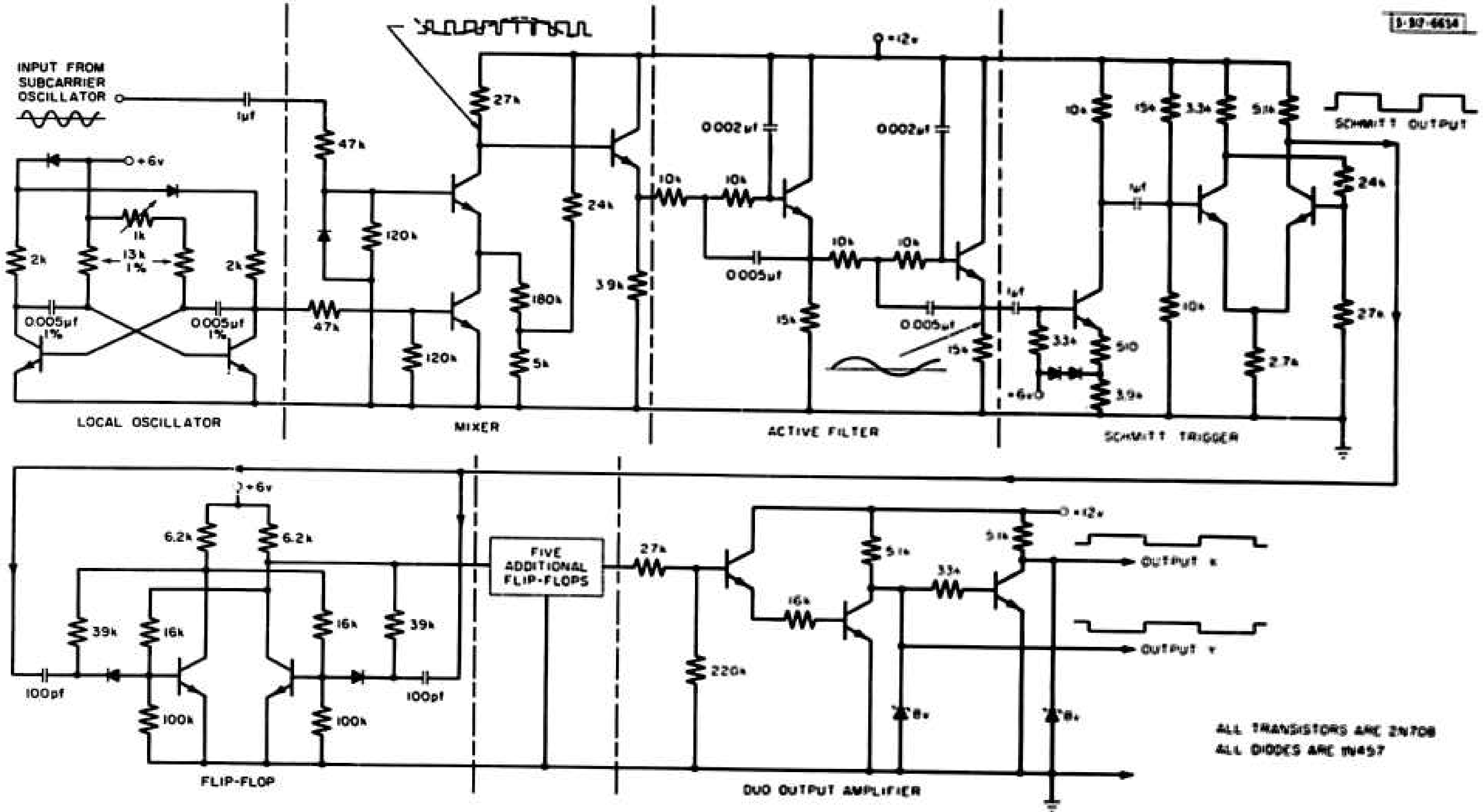


Figure 7. Circuit Diagram of One Channel Encoder.

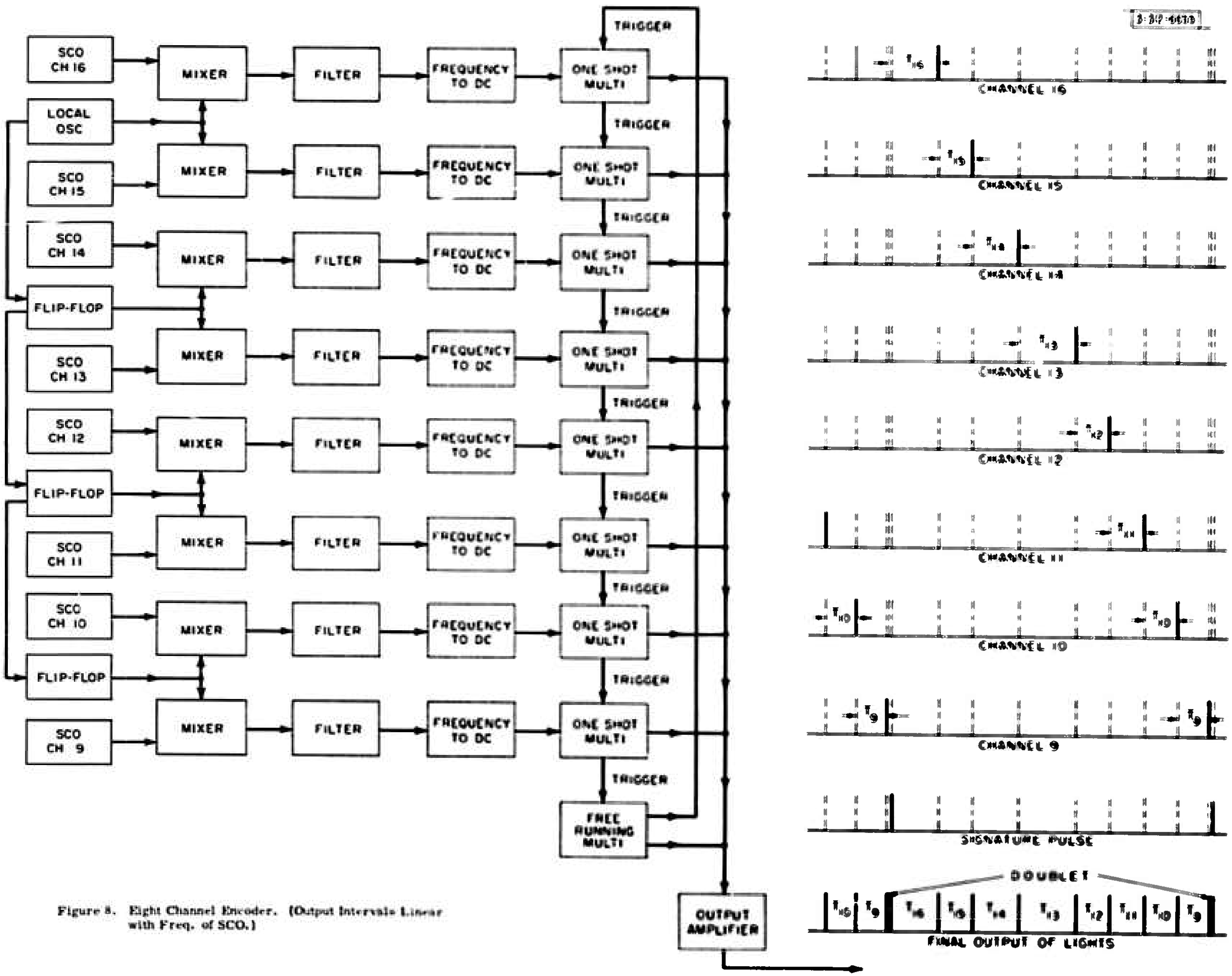


Figure 8. Eight Channel Encoder. (Output Intervals Linear with Freq. of SCO.)

channels is 1.3 with the important EXCEPTION of 1.5 between channels 13 and 14. This value of 1.5 compensated sufficiently for the 1.3 ratios to allow the use of a master oscillator technique on a group of channels centered about channels 13 and 14. The master oscillator approach reduces error because only one oscillator is needed and any error that does result from oscillator drift can be calibrated out if necessary by employing a dummy channel.

Immediately following each filter in the block diagram of Fig. (8) is a block labled "freq. to dc." This block converts the frequency from the filter into a dc voltage proportional to:

$$E_{dc} = A\left(e^{-\frac{\left|f - f_c\right|}{RC}} + B\right) \tag{17}$$

where R, C, A and B are to be determined and $\left|f - f_c\right|$ is the difference frequency. T is the period of the one shot multivibrator which is controlled by the dc voltage E_{dc}. Knowing the range of $\left|f - f_c\right|$ and calculating the desired value for the time constant of the multivibrator, RC, is then determined. B and A are constants determined by the circuit parameters of the monostable multivibrator. E_{dc} is now used to bias the emitter of the non-triggered transistor in the multivibrator. With proper setting of the afore mentioned constants the period of this multivibrator will be linearly dependent on the input frequency and thus linearly dependent on the original data.

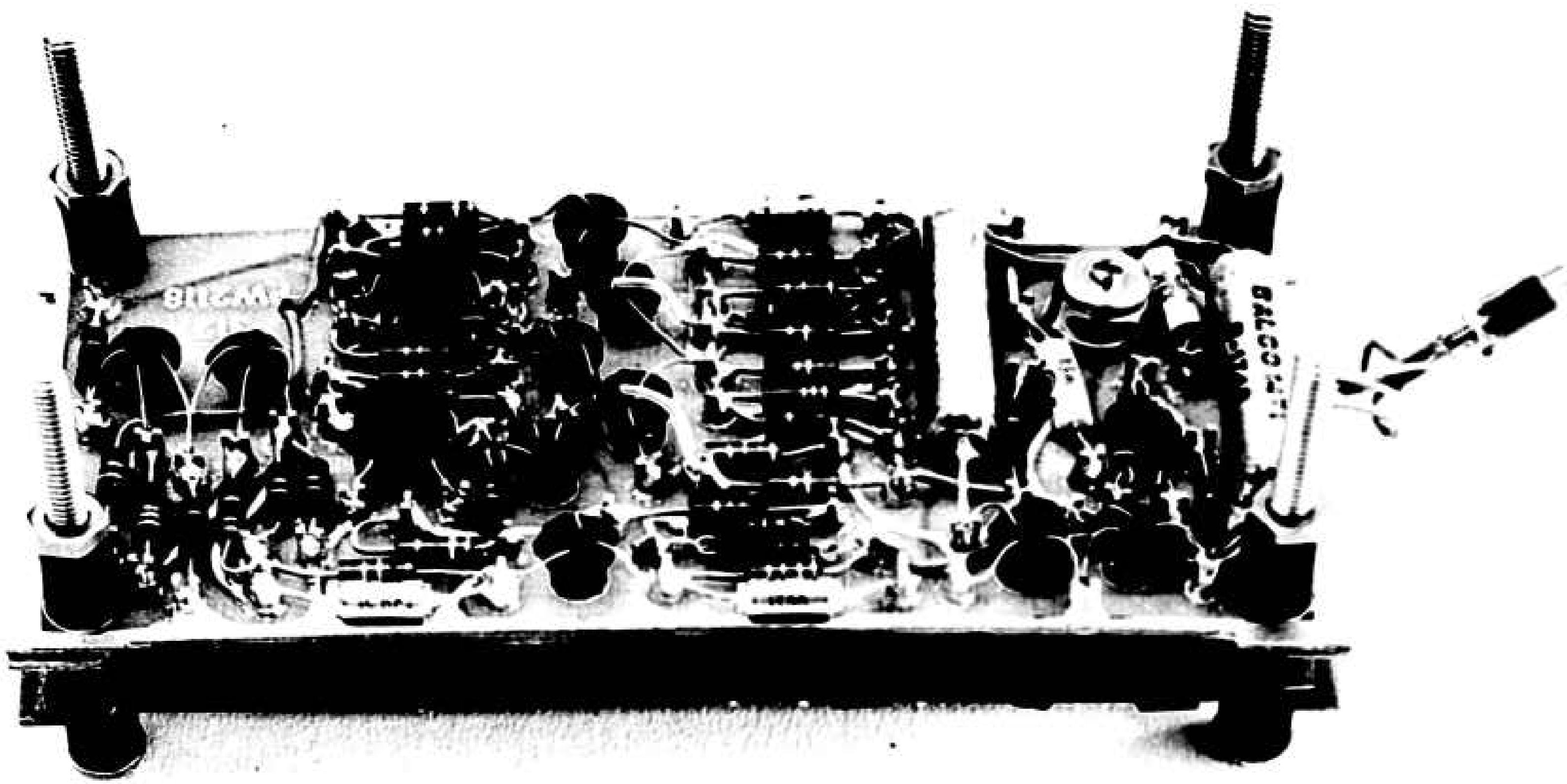
The output pulse of the first multivibrator occuring at the end of T_A is fed through the output amplifier to trigger the flashing light. This same pulse is also used to trigger the next multivibrator so that the beginning of one interval is concurrent with the termination of the previous interval. At the end of a cycle in which all channels have been sampled an unsymmetric multivibrator is triggered resulting in the "doublet" pulse. The doublet or signature pulse is also used to

recycle the multivibrators. This operation is shown in the right side of Fig. 8. The long period of the non-triggered side of this signature multivibrator is larger than the maximum cycle time of the encoder and is employed only to originate initial cycling or to maintain partial cycling in the event that the cycle is broken.

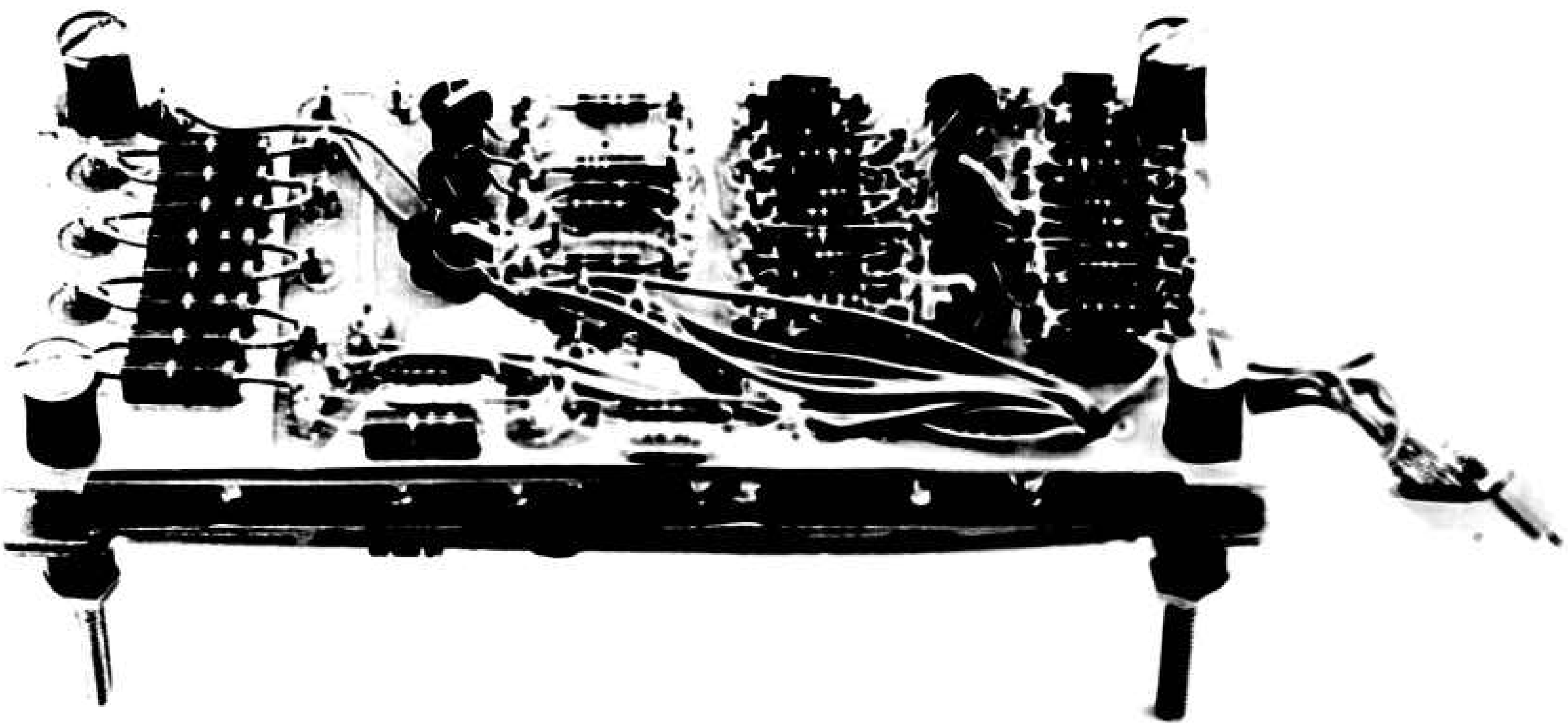
The frequency ranges of the master oscillator as a function of the number of channels needed is shown in Fig. (9). The preferred channels were chosen to maximize the range of the choice in the master oscillator frequency.

No. of Channels	Preferred Channels	Master OSC Frequency fc
8	9-16	35,272 ± 1728 cps
7	9-15	36,796 ± 2052 cps
6	10-15	34,892 ± 3600 cps
5	10-14 (or 10, 11, 13 14, 15)	35,316 ± 4040 cps
4	Any combination of 10, 11, 13, 14, 15 Results in Sufficiently Wide Selection of f_c	
3		35,316 ± cps
2		
1		

Figure 9. SELECTION OF CHANNELS AND FREQUENCY OF MASTER OSCILLATOR



One Channel Encoder (Working Model) Top View.



One Channel Encoder (Working Model) Bottom View.

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